(54) Title: METHOD AND APPARATUS COMPRISING SYNCHRONIZING MEANS FOR PACKETIZING AT LEAST TWO DATA STREAMS

(57) Abstract

There are voice signals from at least two voice sources which are multiplexed by an ATM Adaptation Layer 2 multiplexer (AAL2 MUX) into the same ATM connection after having been processed by a packetizing means such that the AAL2 MUX is presented with assembled AAL2 packets of a suitable periodicity. If the voice data signals are to be compressed, the packetizing means comprises an encoder of a suitable voice codec, e.g. ADPCM, GSM, LD-CELP or CS-ACELP. The voice encoders may also include means for silence removal. The packetizing means further comprises a packetizer, either separated from the voice encoder, or as a part thereof. The packet releases from the packetizing means are synchronized to occur virtually simultaneously. In this way, the released packets from the packetizing means arrive almost simultaneously at the AAL2 MUX, with the result that an ATM cell may be filled with AAL2 packets well before the expiration of the timer Timer-CU of the AAL2 MUX.

B-1

METHOD AND APPARATUS COMPRISING SYNCHRONIZING MEANS FOR PACKETISING AT LEAST TWO DATA STREAMS

Technical Field of the Invention

This invention relates generally to the packetization of data streams. More particularly, this invention relates to packetizing voice signals and transporting the packetized voice signals using Asynchronous Transfer Mode (ATM) after having multiplexed the packetized voice signals from several sources into the same ATM cell.

Description of Related Art

Voice or speech information in existing telecommunication networks, as e.g. Public Switched Telephone Network (PSTN) and Integrated Services Digital Network (ISDN), is normally transported as PCM coded samples. According to ITU-T G.711, PCM stands for Pulse Code Modulation of voice frequencies and means that the basic time slot is 8 bits every 125 µs, giving rise to channels of 64 kbit/s. Consequently, PSTN and ISDN are 64 kbit/s circuit-switched networks. Circuit stands for a set of physical transmission resources, e.g. lines and exchanges, that provide for a two-way transfer of message signals from source to destination in a telecommunication system. Several circuits are time multiplexed over one link by joining together several time slots in a frame which is again repeated with a certain frequency. A circuit will always use the same time slot in the frame during the complete duration of the session. When transporting the PCM coded samples over an ATM network, i.e. a packet switching network instead of a circuit-switched data network, one way for the ATM network to achieve this is to emulate a virtual circuit. The emulation is carried out by a Circuit Emulation Service (CES) as specified by the ATM Forum Circuit Emulation Service Interoperability Specification Version 2.0, which comprises an ATM Adaptation Layer 1 (AAL1) adapting circuits of constant bit rate to an ATM cell transport.

Transporting a 64 kbit/s circuit using AAL1 can be done in basically two ways. One is to entirely fill the ATM cell payload, i.e. the information field, generally of 47 octets

5   (48 octets for the information field - 1 octet for the AAL1 header) with PCM samples,
     resulting in an ATM packetization delay of 47 * 125 μs, i.e. 5875 ms. The extra
     overhead added by the 5 octets ATM cell header and 1 octet AAL1 packet header
     amounts to roughly 13 percent of the nominal bit rate of 64 kbit/s. The other way is to
     only partially fill the ATM cell payload with PCM samples before sending it, and thus

10   reduce the ATM cell packetization delay, but paying penalty by increased overhead.
     For instance, by only filling the cell by a sixth, i.e. by around 8 PCM samples, the
     delay can be reduced to around 1 ms. It is important to keep delay at a low level for all
     parts of the network, since delay in combination with echo may severely degrade the
     perceived voice quality.

15

     Voice signals may also be carried in other encoding formats than the above mentioned
     PCM format. These encoding formats being a part of a voice codec may be according
     to e.g. the Global System for Mobile Telecommunication (GSM) standards, Adaptive
     Differential PCM (ADPCM) standards, Low Delay Code Excited Linear Prediction

20   (LD-CELP) standards or Conjugate Structure Algebraic Code Excited Linear
     Prediction (CS-ACELP) standards. A common property for all these voice codecs is
     that they compress the voice signals according to a compression algorithm such that
     they produce coded voice information, i.e. packets or frames, with a resulting bit rate
     which is lower than the common 64 kbit/s PCM based coding scheme. The result is

25   therefore, that when using AAL1, the ATM packetization delay and /or overhead (if
     partially filled cells are used) increases roughly by a factor equal to the degree of
     compression. For instance, an 8 kbit/s CS-ACELP coded voice signal would result in
     an ATM packetization delay as long as 47 ms. This could as before be traded by
     partially filling the cell, but with the same scaling factor of 8 for the resulting

30   bandwidth overhead. In addition, some voice codecs based on GSM and CS-ACELP
     already have an inherent algorithm delay which is fairly large which also has to be
     added in a total network perspective.

     The different voice codecs produce packets at varying rates and sizes which are not

35   commensurable with the ATM payload length, resulting in padding or segmentation

3

problems. With these voice encoders it is also possible to filter out silent periods of speech and to reduce or stop the emission of voice data during those silent periods (hereinafter referred to as silence removal). Silence removal results in a variable bit rate, which is not suitable for a constant bit rate transport service such as the CES based on AAL1.

A new ATM Adaptation Layer (AAL) for the purpose of transporting packetized low bit rate data, such as packetized voice, and which data is real-time critical and of variable rate and length, is in the process of being standardized. This new AAL is denoted AAL type 2 (AAL2), which is specified in the ITU-T Draft I.363.2, B-ISDN ATM Adaptation Layer Type 2 Specification, Seoul, February 1997. AAL2 is designed to provide support for applications requiring low delay with reasonable bandwidth efficiency and variable packet sizes. AAL2 is also asynchronous.

In AAL2, voice signals from several sources are multiplexed into the same ATM connection. This is done by encapsulating user data into AAL2 packets, which are multiplexed using the AAL2 multiplexer function (AAL2 MUX) inherent in the AAL2 Common Part Sublayer, into an ATM connection. Reduced delay is provided through inclusion of a Combined Use-timer (Timer_CU) in the AAL2 which guarantees a maximum holding time of user data, or AAL2 packets, before transmission of the carrying ATM cell. The maximum packetization delay introduced by the AAL2 ATM is then basically equal to the value of the Timer_CU. The penalty to be paid is overhead in bandwidth if an ATM cell is not completely filled upon transmission.

In a typical scenario using AAL2, there are n voice sources which are multiplexed by an AAL2 MUX into the same ATM connection after having been processed by an encoder with a suitable voice coding algorithm. Each voice encoder may be combined with a packetizer such that the AAL2 MUX is presented with packets of a suitable periodicity. The voice encoders may also include means for silence removal. Hence, after processing, the AAL2 MUX is presented with packets having a variable length p and/or packet rate f. A typical appropriate value for T_{Timer_CU} is 1 ms, which is based on

5   calculations and simulations. Generally, T_{Timer_CU} is much less than the packet

periodicity t (= $1/f$) for most encoders, also when they are combined with packetization

procedures similar to those proposed by ITU-TG.764, Voice Packetization-Packetized

Voice Protocols, or FRF.11, Voice over Frame Relay Implementation Agreement, FR

Forum. The result is that there is a low probability of including several packets from

10   different voice sources in the same ATM cell when the number of sources are few, e.g.

less than 20. The remaining part of the ATM cell, which is not filled with packets at

the expiration of the Timer_CU, is then padded. However, at few sources, the resulting

padding overhead very much destroys the advantage of having employed encoders,

with or without silence removal. A situation where this could be a problem is when

15   doing PBX trunking. PBX stands for Private Branch Exchange and is essentially an

on-premise telephone exchange system that services a number of telephones within a

building.

It should therefore be appreciated that there is a need for a device and related system

20   and method for more efficiently transporting packetized data from several sources on a

packet oriented transport medium and in particular when there are only a few sources.

Summary of the Invention

25   This invention is embodied in a method according to claims 1 to 6, a related apparatus

according to claims 7 to 17 and a related system according to claims 18 to 25.

According to the invention there is an apparatus and a corresponding method for

packetizing at least two data streams, comprising at least two packetizing means, each

30   packetizing means producing first packets from said respective data stream, and a

synchronizing unit for coordinating the packetizing means to substantially

synchronously produce said first packets.

Preferably, the synchronizing unit comprises means for receiving permission requests

35   for packetizing from the respective packetizing means, means for transmitting

5   permission to packetize to the respective packetizing means, and a synchronizing unit
for transmitting said permission to packetize to the respective packetizing means to
coordinate said respective packetizing means to release substantially synchronously
said first packets with other packetizing means already releasing first packets.

10   The packetizing means which virtually simultaneously produce first packets may form
a synchronization group. The synchronizing unit may comprise means for forming
synchronization groups, each synchronization group comprising between one and a
predetermined number of packetizing means. The number of packetizing means
belonging to a synchronizing group depends on the chosen encoding format and on
15   how efficient the designer of the system wishes the system to be.

The packetizing means may either comprise: solely a packetizer for packetizing the
respective data stream into first packets; encoding means for encoding the respective
data stream into data blocks according to a predetermined encoding format, e.g.
20   ADPCM or LD-CELP, and a packetizer for producing first packets from the data
blocks; or a packetizer for producing data blocks from the respective data stream and
encoding means for encoding the data blocks into first packets according to a
predetermined encoding format, e.g. GSM or CS-ACELP. The packetizer may thus be
an integral part of a specified voice encoding scheme.
25

The packetizing means may furthermore comprise means for silence removal.
Providing the silence removal is well known in the art.

According to the invention there is furthermore a system for packetizing at least two
30   data streams, comprising at least two sources producing the respective data streams, an
apparatus for packetizing said at least two data streams, comprising at least two
packetizing means, each packetizing means producing first packets from said
respective data stream, and a synchronizing unit for coordinating the packetizing
means to substantially synchronously produce said first packets, said system further
35   comprising a packet oriented transport medium and multiplexing means for

5   multiplexing the resulting first packets into respective second packets onto the packet
     oriented transport medium.

     The data streams preferably comprises digitized audio and/or video information .

10   The packet oriented transport medium may be a transport medium according to the
     ATM, the second packets are then ATM-cells, and the multiplexing means is an AAL2
     MUX for producing AAL2-packets from respective first packets and multiplexing the
     AAL2-packets into the second packets. Preferably, the predetermined number of
     packetizing means that belong to a synchronization group is determined by the number
15   of AAL2-packets that fit into the respective ATM-cells. As the ATM cell payload size
     is framed, there is a possibility that the number of AAL2 packets that fit into the ATM
     cell payload is not an integer number. In these cases, two possibilities exist for the
     remainder of the ATM cell payload. The first is to pad the cell with dummy
     information. The second is to use AAL2 segmentation, i.e. to segment the last AAL2
20   packet into two ATM cells. Both padding and segmentation are functions which are
     well known to the person skilled in the art. The predetermined number of packetizing
     means belonging to a synchronization group may thus stretch over more than one ATM
     cell. The number is mainly determined by the AAL2 packet size in comparison with
     the ATM cell payload and how efficient one wants to design the system.

25
     Hence, according to a preferred embodiment of the invention, there are voice signals
     from at least two voice sources which are multiplexed by an ATM Adaptation Layer 2
     multiplexer (AAL2 MUX) into the same ATM connection after having been processed
     by a packetizing means such that the AAL2 MUX is presented with assembled AAL2
30   packets of a suitable periodicity. If the voice data signals are to be compressed, the
     packetizing means comprises an encoder of a suitable voice codec, e.g. ADPCM,
     GSM, LD-CELP or CS-ACELP. The voice encoders may also include means for
     silence removal. The packetizing means further comprises a packetizer, either
     separated from the voice encoder, or as a part thereof. The packet releases from the
35   packetizing means are synchronized to occur virtually simultaneously. In this way, the

5   released packets from the packetizing means arrive almost simultaneously at the AAL2

MUX, with the result that an ATM cell may be filled with AAL2 packets well before

the expiration of the timer Timer_CU of the AAL2 MUX.

According to another embodiment of the invention, the packet oriented transport

10   medium may be Frame Relay or IP. The invention may thus be part of a system

providing Voice over Frame Relay or Voice over IP.

An advantage of the invention is that it is possible to efficiently fill the second packets,

e.g. ATM cells, with data from a few sources before the second packets are transported

15   on the packet oriented transport medium. As a result it is possible to maintain the

packetization delay at a low level, but still providing a high level of bandwidth

utilization/efficiency.

The current use of AAL2 for packetized voice transport is optimized for mobile

20   applications where the typical number of sources multiplexed on a single ATM

connection is in the few hundreds. AAL2 has attracted a lot of attention to be used also

for PBX trunking. However, the number of voice sources, e.g. 50 or less, for PBX

trunking is significantly lower than for a mobile trunking application which makes the

use of AAL2 less optimal. However, with the use of the proposed solution, the current

25   problem of few sources can be overcome and AAL2 might prove to be as useful for

PBX trunking as for mobile trunking. Note that PBX trunking is just one example of

where few sources might be present. Another example could be an ATM based

residential access network used to carry telephony traffic.

30   **Brief Description of the Drawings**

Fig. 1 is a block diagram depicting a system comprising synchronizing means that is

suitable for practicing the preferred embodiment of the invention;

Fig. 2 is a block diagram depicting packetizing means formed into synchronization

groups;

5   Fig. 3 is a diagram depicting the timing of the outputting of packets in accordance with

the preferred embodiment of the invention;

Fig. 4a, 4b and 4c show block diagrams depicting where the packet synch signal is

received by the different types of packetizing means.

Fig. 5 shows a comparison between the result of a calculation of the efficiency as a

10   function of the number of sources where a synchronized and an unsynchronized

release of the voice encoders are performed;

Fig. 6 shows a block diagram of the synchronizing unit and the packetizing means

according to a preferred embodiment of the invention;

Fig 7. is a flow diagram according to a preferred embodiment of the invention.

15

Detailed Description of the Embodiments

Fig. 1 is a block diagram depicting synchronizing means 30 that is suitable for

practicing the preferred embodiment of the invention. The synchronizing means 30

comprises a synchronizing unit 32 for synchronizing the activity of a plurality of

20   packetizing means 20 for generating packets having a length p at a rate f. The

synchronization is achieved by using a synchronization bus 34 for transmitting packet

synch signals 36 between the synchronizing unit 32 and the packetizing means 20.

Each packetizing means 20 is connected to a respective source 10 and receives data

input therefrom. The data input may e.g. be voice signals or video signals. The voice

25   signals are preferably a stream of 8 bit PCM coded samples. The video signals may be

low bit rate video according to the ITU-T specification H.261 or H.263. Each

packetizing means 20 is connected to a multiplexer 50 for multiplexing packets

outputted from the packetizing means 20 onto a packet oriented transfer medium 60.

Before outputting packets, the packetizing means 20 transmits an activation request 40

30   to the synchronizing unit 32. The synchronizing unit 32 will then transmit a packet

synch signal 36 via the synchronization bus 34 to the packetizing means 20 for

activating the same in synchrony with other already activated packetizing means 20 or

as a first member of a new synchronization group 2. The term will synchronization

group 2 will be described below.

35

5   According to a preferred embodiment the packet oriented transfer medium is an ATM
       switch network and the multiplexer is an AAL2 multiplexer 50 comprising a
       Timer_CU 51 for guaranteeing a maximum holding time of AAL2 packets, before
       transmission of the carrying ATM cell.

10   Fig. 2 is a block diagram depicting seven packetizing means 20 formed into
       synchronization groups A,B,C by the synchronizing unit 32. Supposing that there
       should be three packetizing means 20 per synchronization group, group A is full and
       group B and C each have room for one more packetizing means 20, The
       synchronization groups are distributed in time to avoid ATM cell bursts.

15
       Fig.3 is a diagram depicting the timing of the synchronous release of packets 3 in
       accordance with the preferred embodiment of the invention. Sources 10 outputs
       independently of each other a stream x,y,z of PCM samples at a rate 64 kbit/s, i.e. 8
       bits every 125 µs. When the stream x,y,z of PCM samples are inputted to the

20   packetizing means 20, the packetizing means 20 requests an activation signal 36 from
       the synchronizing unit 32 that it may begin to generate packets 3. The synchronizing
       unit 32 checks if there are other packetizing means 20 which are already generating
       packets 3. A predetermined number of packetizing means 20 which generates packets 3
       in synchrony is called a synchronization group 2. Either the packetizing means 20 is

25   allocated to an already existing synchronization group 2 or it is allocated to a, by the
       synchronizing unit 32, newly formed synchronization group 2. Until the packetizing
       means 20 has received a packet synch signal 36, it is not allowed to generate packets 3.
       Consequently, until activation, there will be some PCM samples which may be lost.
       The packetizing means 20 will output packets 3 every t ms, where t ms is much larger

30   than 125 µs. The packets 3 are thereafter processed by an AAL2 MUX 50 which
       adapts the packets 3 such that they fit into an ATM cell 4. The ATM cell 4 comprises a
       header 5 and a predetermined of number of AAL2 packets 61. The predetermined
       number of AAL2 packets 61 which fit into an ATM cell 4 depends on the size of the
       packets 3 generated by the packetizing means 20. Since the number of AAL2 packets 3

35   which fit into an ATM cell 4 depends on the size of the packets 3 generated by the

10

5　packetizing means 20, consequently the size of the synchronization groups 2 also
depends on the size of the packets 3.

That the packetizing means 20 are substantially synchronized means that they are
coordinated to release packets such that ATM cells are filled in the most optimal way
10　before the expiration of the Timer_CU of the AAL2 MUX.

For example, as the sample rate is 8 kHz for ordinary PCM coded voice, the AAL2
packet release cannot be made exactly simultaneously, but rather separated by a
multiple of approximately 4 μs, which is the octet arrival time for the PCM samples on
15　a 2 Mbit/s link for samples belonging to different speech channels. This works, since
the sample interval of 125 μs is much less than the resulting voice packet periodicity t,
which is in milliseconds, and that the voice packet periodicity is significantly larger
than the value of Timer_CU 51.

20　Fig. 4a, 4b and 4c show block diagrams depicting where the packet synch signal 36 is
received by the different types of packetizing means 20.The key lies in timing when to
create the blocks used as input to e.g. CS-ACELP encoders, or when to create the
blocks from e.g. an ADPCM encoder, i.e. synchronizing the PCM samples/ bit blocks.
Basically two different types of encoders exist: one which inherently produces packets,
25　e.g. CS-ACELP and GSM encoders; and the other which produces bit blocks with 8
kHz sample rate, e.g. ADPCM and obviously PCM encoders.

Fig 4a shows packetized PCM coded voice;
　　　　　Input: 8 bit PCM samples, 8 kHz sampling rate;
30　output: packets of length p, packet rate of f.

Fig 4b shows packetized ADPCM coded voice;
input : 8 bit PCM samples, 8 kHz sampling rate
output: packets of length p, packet rate of f.

35

5   Fig 4c shows packetized CS-ACELP coded voice;

input: 8 bit PCM samples, 8 kHz sampling rate;

output: packets length of 10 octets, packet rate of 100Hz.

For uncompressed voice (Fig 4a), the packetizing means 20 only comprises a

10   packetizer 21, which then receives packet synch signals 36 from the synchronization

means 32. The packetizer 21, e.g. according to ITU-T G.764, receives 8 bit PCM

samples at a sample rate of 8 kHz and outputs packets of length p at a rate f.

Consequently, packetized PCM coded voice is outputted from the packetizing means

20.

15

An alternative to ITU-T G.764 could be similar to the packetizing method described in

FRF.11, which in substance is based on the same underlying principles as G.764. Yet

another alternative could be according to the coming ITU-T recommendation

I.TRUNK which is currently being prepared by the ITU-T, to, among other things,

20   provide methods for packetizing voice data streams within the AAL2 Service Specific

Convergence Sublayer (AAL2 SSCS).

For compressed voice according to the ADPCM standards (Fig. 4b), the packetizing

means 20 comprises a packetizer 21, which receives packet synch signals 36 from the

25   synchronization means 32, and an encoder 22. For an ADPCM encoder 22, a

packetizer 21 has to be added after the encoder in order to produce voice packets

having a suitable length p and rate f, since the ADPCM encoder outputs bit blocks of

5, 4, 3 or 2 bits at a sample rate of 8 kHz. This can be done in accordance with ITU-T

G.764, FRF.11 or I.TRUNK. Consequently, packetized ADPCM coded voice is

30   outputted from the packetizing means 20. ·

An alternative could be an LD-CELP encoder which produces bit blocks of 10 bits,

with a bit rate of 1.6 kHz. Since these bit blocks are too short to be packetized into an

AAL2 packet, a packetizer 21 is added after the LD-CELP encoder.

35

12

5   The same packetizing means structure for obtaining ADPCM or LD-CELP packetized
       voice could be used for packetizing PCM coded voice. This is achieved in that the
       encoder 22 is a null encoder, well known in the art, or comprises means for silence
       removal, which is well known in the art.

10   For compressed voice according to e.g. the CS-ACELP standards (Fig. 4c), the
       packetizing means 20 comprises a CS-ACELP encoder according to ITU-T G.729. A
       CS-ACELP encoder comprises a packetizer 21, which then receives the packet synch
       signals 36 from the synchronization means 32, and an encoder 22. The packetizer 21
       of the CS-ACELP encoder 20 receives 8 bit PCM samples at the sample rate of 8 kHz
15   and outputs 80 octet blocks at the block rate of 100 Hz which are inputted to the
       encoder 22. The encoder 22 then outputs packets of length 10 octets and packet rate of
       100 Hz.

       As explained it is crucial to synchronize the packet release from the packetizing means
20   belonging to the same synchronization group to occur virtually simultaneously. In this
       way, the few AAL2 packets needed to fill the ATM cell arrive at the same time at the
       AAL2 multiplexer. As seen above, an AAL2 packet payload size may be 10 octets
       when using CS-ACELP. With an AAL2 overhead of 3 octets, the overhead being the
       common part sublayer protocol control information (CPS PCI), an ATM cell can
25   contain $47/13 \approx 3 - 4$ AAL2 packets. It is, hence, enough to synchronize the release of
       packets from the 3 - 4 voice encoders in order to fill the ATM cell and achieve
       virtually the maximum efficiency. If e.g. three AAL2 packets are used to fill an ATM
       cell, the remaining unfilled part of the ATM cell is padded. If, on the hand, four AAL2
       packets are used to fill an ATM cell, one AAL2 packet is segmented and put into a
30   subsequent ATM cell. Consequently, it is the role of the synchronizing unit 32 to
       achieve the most optimal filling of the ATM cells by combining synchronization of the
       packetizing means 20 according to the invention, with segmentation and padding,
       which are both well known in the art. What is to be considered as the most optimal
       filling of the ATM cells is decided by the designer of the system, and may depend on
35   available bandwidth, the exact packet length, typical number of sources, etc..

5   Fig. 5 shows the result of a calculation of the efficiency as a function of the number of

sources where an unsynchronized and a synchronized release of the voice encoders is

performed. The voice packet length=10 octets, the packet periodicity=10 ms and the

nominal handwidth per source= 8 kbit/s. Curve a shows utilization without

10   synchronization, curve b utilization with synchronized AA2 packets, and curve c ideal

utilization (8 kbit(s; 64 kbit/s). Curve b is due to that AAL2 packets being segmented

into two ATM cells if an integer number of AAL2 packets does not fit entirely into a

particular ATM cell. In this example the characteristics of CS-ACELP is used whereby

a maximum of three complete AAL2 packets fit into the ATM cell payload. If four

15   AAL2 packets are used to fill an ATM cell, one AAL2 packet is segmented and its

remainder put into a subsequent ATM cell.

Fig. 6 shows a block diagram of the synchronizing means 30 according to a preferred

embodiment of the invention. Accordingly, the synchronizing unit 32 comprises a

20   memory 31, a processor 33 and means 35 for transporting information between the

memory 31 and processor 33 for implementing the synchronization algorithm as shown

in Fig. 7 below. The packetizing means comprises a memory 24, a processor 25, and

and means 26 for transporting information between the memory 24 and processor 25

for implementing the packetization of data. A bus 34 transfers requests for permission

25   to packetize, permissions and return permissions between the processors 26,33 of the

synchronizing unit 32 and several packetizing means 20 respectively via the

communication means 23,37.

The purpose of the request signal 40 is to notify the synchronizing unit 32 of an

30   initialization of a packetizing means 20 which is ready to transmit voice packets. An

initialization takes place when a packetizing means 20 is about to be put in use for an

active voice channel, either when establishing a voice channel or when a voice channel

already is established, e.g when using silence techniques, but at the beginning of a talk

spurt, i.e. after a silence period. It will also notify the synchronizing unit 32 when a

5   packetizing means 20 has become idle, i.e. not in use for a voice channel or at the

beginning of a silence period.

The packet synchronization bus 34 is used to transmit packet synchronization

information from the synchronizing unit 32 to all packetizing means 20. Packet

10   synchronization consists of information at which particular sample a block shall be

initiated to be assembled as input to an encoder 22 or when a bit block coming as

output of an encoder 22 shall be initiated to be packetized in a packetizer 21. Packet

synchronization information needs only to be transmitted once to a particular

packetizing means 20 during the packetizing means 20 active interval, unless

15   resynchronization is needed, e.g. after a silence period. An active interval is the

interval during which the packetizing means 20 is used to packetize / encode an active

voice channel, e.g. during a talk spurt, if silence removal is employed, or during the

full duration of the connection.

20   The synchronizing unit 32 determines, based on knowledge about active packetizers

21, how to synchronize the active packetizer 21. All the packetizers 21 that are

synchronized belong to the same synchronization group. The number of packetizers 21

to be put in a synchronization group is determined by the number of AAL2 packets 61

that fit into one ATM cell 4. The padding shall be minimized or removed entirely by

25   completely filling the ATM cell by means of segmentation. However, to have all

packetizers 21 to simultaneously release their packets will result in a queue of ATM

cells to be transmitted into the ATM connection, which will lead to additional delay

due to the limited rate of the ATM connection and the cell queue. Packetizers 21 shall

therefore be distributed in synchronization groups which in turn are evenly distributed

30   in time. However, by synchronizing the voice packets from several sources 10, an

ATM cell 4 will always be filled virtually instantaneously with the effect that the

Timer_CU 51 will not expire. This in fact leads to a reduction of the average

packetization delay.

35   The synchronizing unit 32 preferably takes the following actions:

5   1   At an activation request from an packetizer 21 / encoder 22, it will be allocated to
               an appropriate synchronization group. The rationale is to keep the number of
               synchronization groups low by maximizing the filling of the ATM cells 4.

       2.   Synchronization groups will be created with an even distribution in time to avoid
               ATM cell bursts.

10   3.   Optionally, it shall be possible to reallocate an packetizer 21 / encoder 22 to
               another synchronization group in order to maximize the ATM cell filling. This
               situation is possible when other packetizers 21 / encoders 22 become inactive,
               e.g. due to silence periods. A result will, however, possibly be a loss of one or
               several samples, but not as much as a voice packet, due to the resynchronization

15   process. However, since the number of sources 10 is low, the rate at which
               resynchronization for a particular packetizer 21 / encoder 22 occurs should be
               extremely low.

       Fig 7. is a flow diagram according to a preferred embodiment of the invention and
20   accordingly, in step 100 a packetizer 21 waits until it in step 102 receives a data input
       x,y,z. In step 104 the packetizer 21 determines if it is silence. If the answer is yes, in
       step 106, the packetizer 21 determines if it has a permission to produce first packets. If
       the answer is no, in step 108, the packetizer 21 waits. If the answer is yes, in step 110,
       the packetizer 21 transmits the request to return the permission to the synchronizing
25   unit 32. In step 112 the synchronizing unit 32 receives the request to return the
       permission to packetize and in step 114 removes the packetizer 21 from the
       synchronization group and returns the synchronization group if it is empty to the
       synchronizing unit 32 itself, e.g. to an idle list of unassigned synchronization groups.
       In step 116 the synchronizing unit 32 acknowledges that the permission to packetize
30   has been returned to the synchronizing unit 32 and in step 118, the packetizer 21
       receives the permission returned and in step 120 waits. If, in step 104, the packetizer
       21 determines that there is no silence, in step 122, it determines if it has a permission
       to produce first packets. If the answer is no, in step 124, the packetizer 21 requests
       permission to packetize 40 from the synchronizing unit 32. In step 126 the
35   synchronizing unit 32 receives the request for permission 40 and in step 128

16

determines if there is an idle slot in any synchonization group. If the answer is no, in

step 130, the synchronizing unit 32 determines a slot time for a new synchronization

group and establishes the synchronization group. In step 132, the synchronizing unit 32

allocates the request for permission to the synchronization group. If the answer is yes

in step 128, it directly allocates the request for permission to the synchronization group

in step 132. In step 134 the synchonizing unit 32 transmits permission to packetize to

the packetizer 21, and in step 136 the packetizer 21 receives the permission. In step

138 the packetizer 21 puts the input data in a block and if the block is full, sends the

block as first packet at the allocated time slot defined by the synchronization group. If

the answer in step 122 is yes, the packetizer 21 directly puts the input data in a block

and if the block is full, sends the block as first packet at the allocated time slot defined

by the synchronization group as in step 138. In step 140 the packetizer 21 waits.

5   Claims

1.   Method of packetizing at least two data streams comprising the step of producing

first packets from the respective data stream, characterized by further comprising the

step of coordinating the packetizing of the data streams to substantially synchronize

10   the first packets resulting from the respective data streams.

2. Method according to claim 1, wherein the coordinating step comprises the steps of

- a respective packetizing means receiving the respective data streams,

- the respective packetizing means requesting permission for packetizing from a

15   synchronizing means,

- the synchronizing means receiving the permission request from the respective

    packetizing means to packetize first packets,

- the synchronizing means transmitting permission to packetize to the respective

    packetizing means, to coordinate said respective packetizing means to release

20   substantially synchronously said first packets with other packetizing means already

    releasing first packets.

3.   Method according to claim 2, wherein the step of requesting permission is

performed when setting up a data channel or upon arrival of a data spurt.

25

4.   Method according to any one of the preceding claims, further comprising the

steps of

- the synchronizing means forming synchronization groups, each synchronization

    group comprising between one and a predetermined number of packetizing means

30   and the packetizing means belonging to each synchronization group substantially

    synchronously releasing first packets,

- the synchronizing means when receiving said permission request from the respective

    packetizing means, determining if any synchronization group has been allocated less

    than the predetermined number of packetizing means and allocating said respective

5   packetizing means to such a synchronization group, and if not then forming a new

synchronization group.

5.   Method according to any one of the preceding claims, further comprising the step

of multiplexing the resulting first packets into respective second packets onto a packet

10   oriented transport medium.

6.   Method according to claim 5 when depending on claim 4, comprising the step of

the synchronizing means selecting said predetermined number of packetizing means

belonging to each synchronization group, depending on how many first packets fit into

15   said second packet.

7.   Apparatus for packetizing at least two data streams (x,y,z), comprising at least

two packetizing means (20), each packetizing means (20) producing first packets (3)

from said respective data stream (x,y,z), characterized by further comprising

20   synchronizing means (30,32,34,36,40) for coordinating the packetizing means (20) to

substantially synchronously produce said first packets (3).

8.   Apparatus according to claim 7, wherein the synchronizing means (30) comprises

means (32,34) for receiving permission requests for packetizing (40) from the

25   respective packetizing means (20), means (32,34) for transmitting permission to

packetize (36) to the respective packetizing means, and a synchronizing unit (32) for

transmitting said permission to packetize to the respective packetizing means to

coordinate said respective packetizing means to release substantially synchronously

said first packets with other packetizing means already releasing first packets.

30

9.   Apparatus according to claim 8, wherein the synchronizing unit (32) comprises

means for forming synchronization groups (2), each synchronization group (2)

comprising between one and a predetermined number of packetizing means (20).

5   10.   Apparatus according to any one of claims 7 to 9, wherein the packetizing means

(20) comprises a packetizer (21) for packetizing the respective data stream into first

packets (3).

11.   Apparatus according to any one of claims 7 to 9, wherein the packetizing means

10   (20) comprises encoding means (22) for encoding, according to a predetermined

encoding format, the respective data stream (x,y,z) into data blocks and a packetizer

(21) for producing first packets (3) from the data blocks.

12.   Apparatus according to claim 11, wherein the predetermined encoding format is

15   Adaptive Differential Pulse Code Modulation (ADPCM) or Pulse Code Modulation

(PCM) with silence removal.

13.   Apparatus according to any one of claims 10 to 12, wherein the packetizer (21) is

the packetizing protocol ITU-T G.764 or any derivatives therof.

20

14.   Apparatus according to any one of claims 7 to 9, wherein the packetizing means

(20) comprises a packetizer (21) for producing data blocks from the respective data

stream and encoding means (22) for encoding, according to a predetermined encoding

format, the data blocks into first packets (3).

25

15.   Apparatus according to claim 14, wherein the predetermined encoding format is

Global System for Mobile communication (GSM), Coding of Speech at 16 kbit/s Using

Low-Delay Code Excited Linear Prediction (LD-CELP) or Coding of Speech at 8

kbit/s Using Conjugate Structure Algebraic Code Excited Linear Prediction (CS-

30   ACELP).

16.   Apparatus according to any of claims 10 to 15, wherein the packetizer (21)

comprises means (23) for requesting permission for packetizing and means (24) for

receiving said permission to packetize.

35

5   17.   Apparatus according to any of claims 11 to 16, wherein the packetizing means
(20) comprises means for silence removal.

18.   System for packetizing at least two data streams, comprising at least two sources
(10) producing the respective data streams, a device according to any one of claims 6

10   to 16, a packet oriented transport medium (60) and multiplexing means (4,5,50,51,61)
for multiplexing the resulting first packets (3) into respective second packets (4) onto
the packet oriented transport medium (60).

19.   System according to claim 18, wherein the data streams comprises digitized audio

15   and/or video information.

20.   System according to claim 18 or 19, wherein the data streams comprises voice
data having 8 bit PCM samples at 8 kHz sample rate.

20   21.   System according to any of claims 18 to 20, wherein the packet oriented transport
medium (60) is a transport medium according to the Asynchronous Transfer Mode
(ATM), the second packets (4) are ATM-cells, and the multiplexing means (50) is an
AAL2-multiplexer (50) for producing AAL2-packets (61) from respective first packets
(3) and multiplexing the AAL2-packets into the second packets.

25

22.   System according to claim 21, wherein the predetermined number of packetizing
means (20) that belong to a synchronization group (2) is determined by the number of
whole AAL2-packets (61) that fit into the respective ATM-cells.

30   23.   System according to claim 21, wherein the predetermined number of packetizing
means (20) that belong to a synchronization group (2) is determined by the number of
AAL2-packets (61) that optimally fit into the respective ATM-cells, when the AAL2-
packets are segmented and the ATM-cells padded.

5   24.   System according to any of claims 18 to 20, wherein the packet oriented transport medium (60) is a transport medium according to Frame Relay or IP.

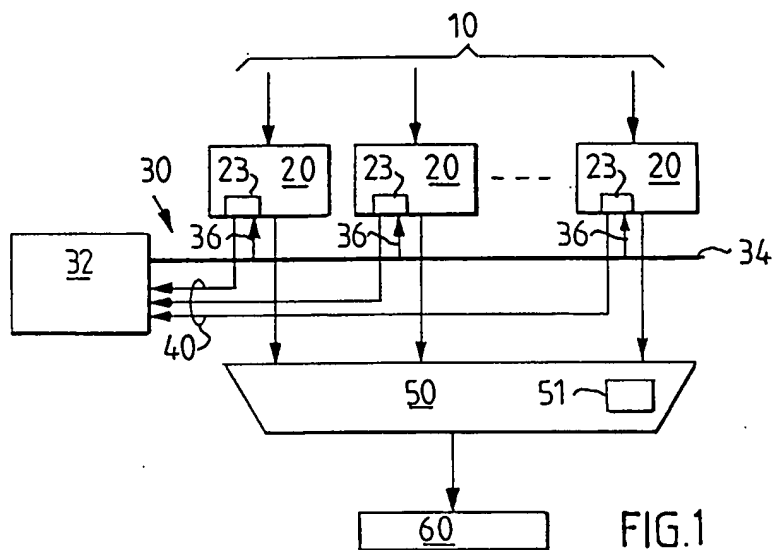25.   System according to any on of claims 18 to 24, wherein the system comprises less then 50 sources.

1/4

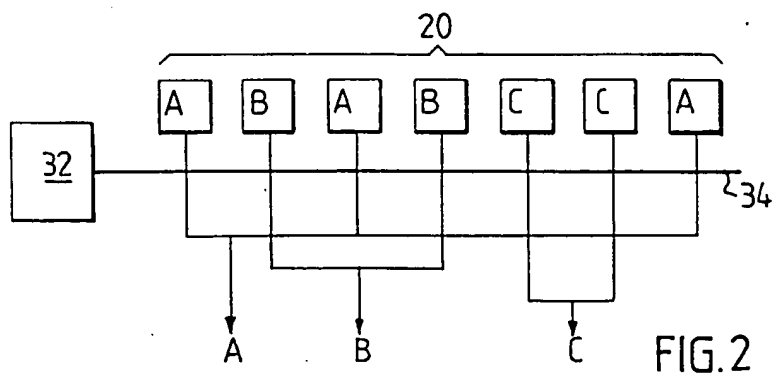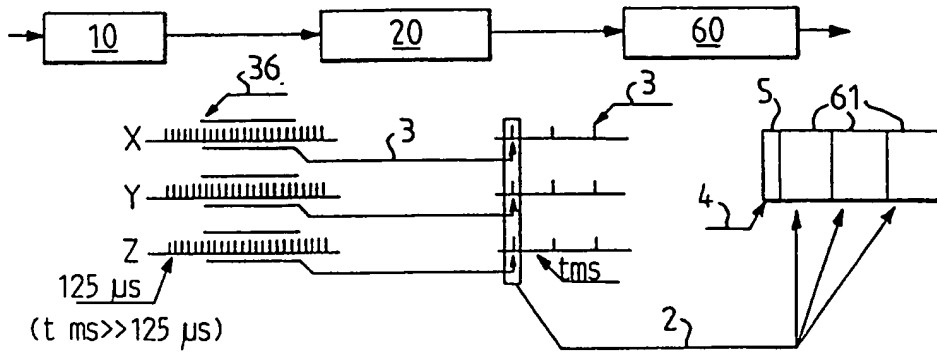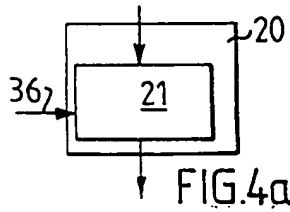FIG.1

FIG.2

FIG.3

FIG.4a
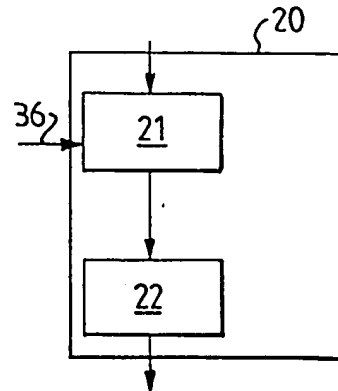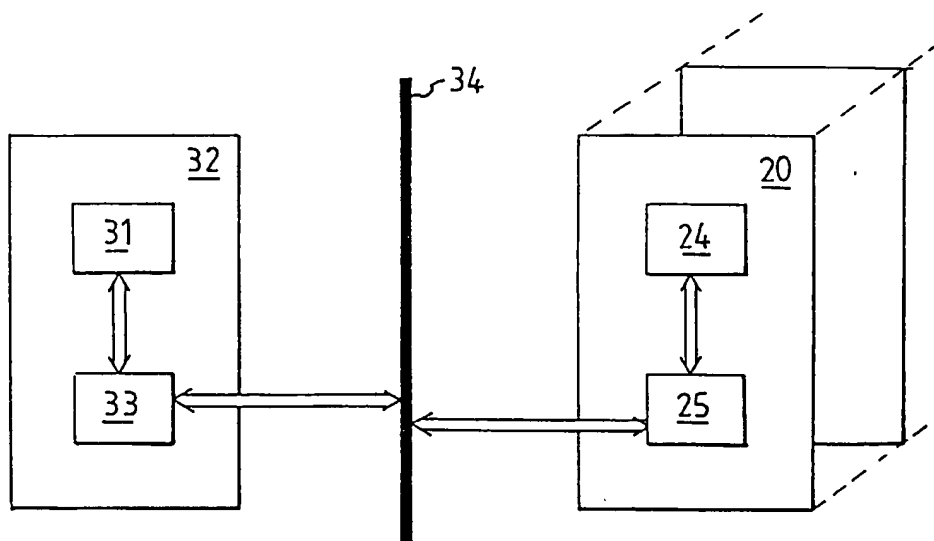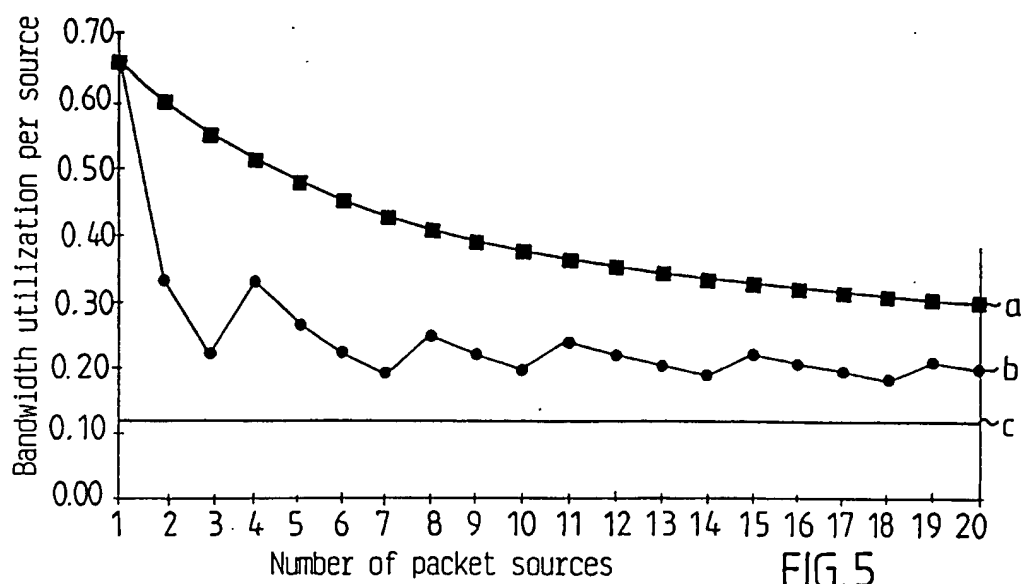

FIG.4b

FIG.4c

FIG.5
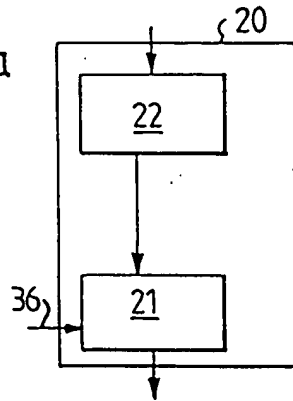
FIG.6

INTERNATIONAL SEARCH REPORT

	International application No. PCT/SE 98/01591

A. CLASSIFICATION OF SUBJECT MATTER

IPC6: H04Q 11/04, H04L 12/56

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

IPC6: H04Q, H04L

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

SE,DK,FI,NO classes as above

Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	US 5327421 A (THOMAS L. HILLER ET AL), 5 July 1994 (05.07.94), column 1, line 52 - column 2, line 11; column 2, line 46 - column 3, line 57, claims 1-9, abstract --	1-25
A	US 5623491 A (FREDERICK H. SKOOG), 22 April 1997 (22.04.97), see whole document --	1-25
A	US 5322882 A (THOMAS L. HILLER ET AL), 6 June 1995 (06.06.95), claims 1-4, abstract --	1-25

X	Further documents are listed in the continuation of Box C.	X	See patent family annex.

*	Special categories of cited documents:	"T"	later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention
"A"	document defining the general state of the art which is not considered to be of particular relevance		
"E"	erlier document but published on or after the international filing date	"X"	document of particular relevance: the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone
"L"	document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)	"Y"	document of particular relevance: the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art
"O"	document referring to an oral disclosure, use, exhibition or other means		
"P"	document published prior to the international filing date but later than the priority date claimed	"&"	document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
6 April 1999	1 3 -04- 1999

Name and mailing address of the ISA/	Authorized officer
Swedish Patent Office Box 5055, S-102 42 STOCKHOLM Facsimile No. +46 8 666 02 86	Ewa Kowalska Telephone No. +46 8 782 25 00

Form PCT/ISA/210 (second sheet) (July 1992)

4/4

(100) Wait

102 Input data

104 Is it silence?

No

Yes

106 Does the packe-tizer have a permission to send 1st packets

No

Yes

(108) Wait

110 Return per-mission

112

114 Remove the packetizer from the synch. group and return synch. group if it is empty

116

118 Per-mission returned

Does the packe-tizer have a per-mission to send 1st packets?

(120) Wait

122

Yes

No

124 Request permission

126 Idle slot in any synch. group?

128

No

Yes

130 Determine slot time for new group

132 Allocate the request to the synch. group

134

136 Permission

138 Put data in block and if the block is full, send the block as 1st packet at the allocated time slot

(140) Wait

Packetizer

Synchronization Unit
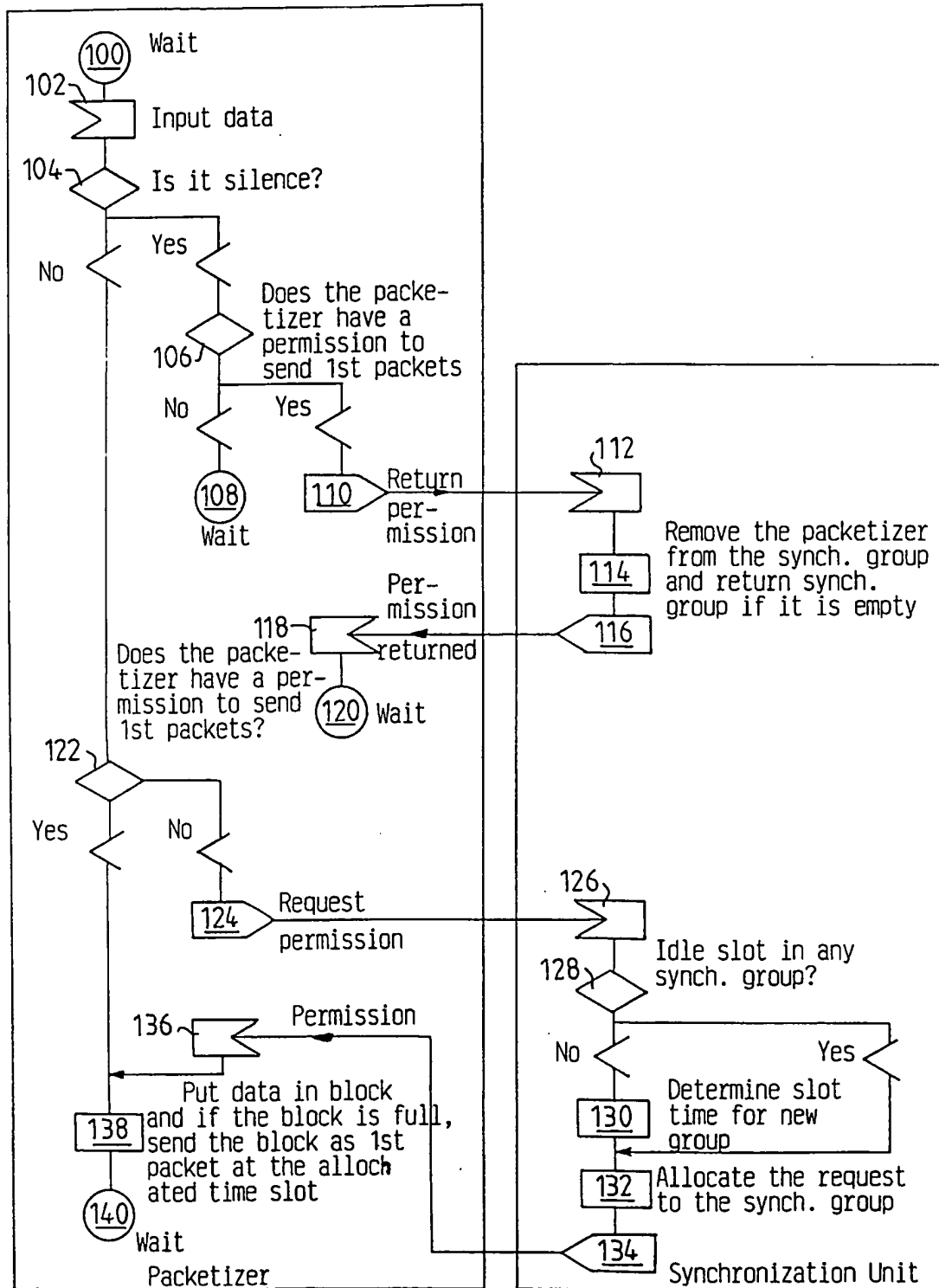
FIG.7

INTERNATIONAL SEARCH REPORT

International application No.

PCT/SE 98/01591

C (Continuation). DOCUMENTS CONSIDERED TO BE RELEVANT		
Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	GB 2291561 A (NORTHERN TELECOM LIMITED), 24 January 1996 (24.01.96), figure 4, claim 1, abstract -- ---------	1-25

Patent document cited in search report	Publication date	Patent family member(s)	Publication date
US 5327421 A	05/07/94	AT 153203 T	15/05/97
		AU 671449 B	29/08/96
		AU 3923193 A	08/06/94
		CA 2148356 A	26/05/94
		CZ 284040 B	15/07/98
		CZ 9501160 A	14/02/96
		DE 69310762 D,T	16/10/97
		EP 0669067 A,B	30/08/95
		SE 0669067 T3	
		ES 2101307 T	01/07/97
		FI 952174 A	05/05/95
		HU 71152 A	28/11/95
		HU 9501305 D	00/00/00
		JP 8503344 T	09/04/96
		MX 9306775 A	31/01/95
		NO 951760 A	05/05/95
		NZ 251551 A	28/08/95
		PL 172499 B	30/09/97
		PL 308639 A	21/08/95
		SG 43781 A	14/11/97
		WO 9411975 A	26/05/94
US 5623491 A	22/04/97	AU 5317396 A	08/10/96
		CA 2214128 A	26/09/96
		EP 0815703 A	07/01/98
		JP 10511245 T	27/10/98
		WO 9629840 A	26/09/96
US 5322882 A	06/06/95	DE 68925952 D,T	25/07/96
		EP 0376052 A,B	04/07/90
		JP 2173061 A	04/07/90
		JP 6008386 B	02/02/94
		KR 9601227 B	24/01/96
GB 2291561 A	24/01/96	DE 69504337 D,T	07/01/99
		EP 0761073 A,B	12/03/97
		SE 0761073 T3	
		GB 9410294 D	00/00/00
		JP 10500545 T	13/01/98
		WO 9532597 A	30/11/95